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(71) Applicant: MOTOROLA, INC. [US/US]; 1303 East Algonquin Road, Schaumubrg, IL 60196 (US).

(72) Inventors: XIE, Qiaobing,; 1509 N. Smith Road, Apt. 204, Palatine, IL 60067 (US). GUPTA, Sanjay,; 9565 Nicklaus Lane, Lakewood, IL 60014 (US).

(74) Agents: WILLIAMS, Lalita P., et al.; MOTOROLA, INC., Intellectual Property Dept., 1303 East Algonquin Road, Schaumburg, IL 60196 (US).

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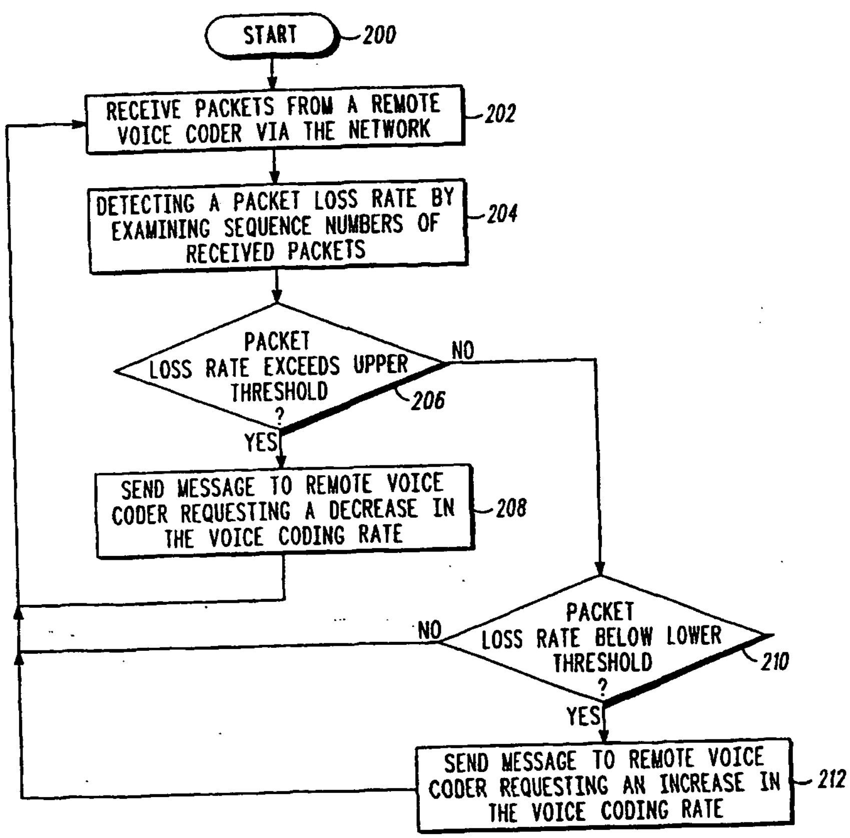
(54) Title: METHOD AND SYSTEM FOR RATE ADAPTATION IN A PACKET VOICE SYSTEM



(57) Abstract: In a network, transmitted voice quality is improved by receiving voice packets from a remote voice coder, wherein the received voice packets have been encoded at a higher encoding rate. If a packet loss rate that exceeds a threshold loss rate is detected, a reduce encoding rate message is sent to the remote voice coder to cause the remote voice coder to encode future voice packets at a lower encoding rate. The voice packets may include a change encoding rate message in an error tolerant portion of the voice packet. The change encoding rate message may also include parity information for verifying the integrity of the message.

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INTERNATIONAL SEARCH REPORT

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B. FIELDS SEARCHED
Minimum documentation searched (classification system followed by classification symbols)
IPC 7 G10L H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

EPO-Internal

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	WO 99 31838 A (STEWART JOHN SIDNEY ;RAMASWAMY KUMAR (US); KNUTSON PAUL GOTHARD (U) 24 June 1999 (1999-06-24) page 9, line 26 -page 10, line 15 ---	1,7
X	GUERRI J C ET AL: "A feedback packet-level error control for real-time applications in wireless networks" VEHICULAR TECHNOLOGY CONFERENCE, 1999. VTC 1999 - FALL. IEEE VTS 50TH AMSTERDAM, NETHERLANDS 19-22 SEPT. 1999, PISCATAWAY, NJ, USA,IEEE, US, 19 September 1999 (1999-09-19), pages 879-883, XP010353105 ISBN: 0-7803-5435-4 paragraph '0003! ---	1,7

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[X] Further documents are listed in the continuation of box C.

[X] Patent family members are listed in annex.

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NL - 2280 HV Rijswijk
Tel. (+31-70) 340-2040, Tx. 31 651 epo nl,
Fax: (+31-70) 340-3016

Authorized officer
Krembel, L

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INTERNATIONAL SEARCH REPORT

International Application No
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C.(Continuation) DOCUMENTS CONSIDERED TO BE RELEVANT

Category °	Citation of document, with indication,where appropriate, of the relevant passages	Relevant to claim No.
X	PAKSOY E ET AL: "AN ADAPTIVE MULTI-RATE SPEECH CODER FOR DIGITAL CELLULAR TELEPHONY" 1999 IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING. PHOENIX, AZ, MARCH 15 - 19, 1999, IEEE INTERNATIONAL CONFERENCE ON ACOUSTICS, SPEECH, AND SIGNAL PROCESSING (ICASSP), NEW YORK, NY: IEEE, US, vol. 1, 15 March 1999 (1999-03-15), pages 193-196, XP000898292 ISBN: 0-7803-5042-1 paragraph ‘0005!; figure 1 ---	1,7
A	BRUHN S ET AL: "Concepts and solutions for link adaptation and inband signaling for the GSM AMR speech coding standard" VEHICULAR TECHNOLOGY CONFERENCE, 1999 IEEE 49TH HOUSTON, TX, USA 16-20 MAY 1999, PISCATAWAY, NJ, USA,IEEE, US, 16 May 1999 (1999-05-16), pages 2451-2455, XP010342317 ISBN: 0-7803-5565-2 paragraph ‘00IV! -----	2,3,8,9

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International Application No
PCT/U[illegible]/42468

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WO 9931838 A	24-06-1999	AU 9041998 A	05-07-1999
		CN 1285101 T	21-02-2001
		EP 1040612 A1	04-10-2000
		WO 9931838 A1	24-06-1999

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Published:
— *without international search report and to be republished upon receipt of that report*

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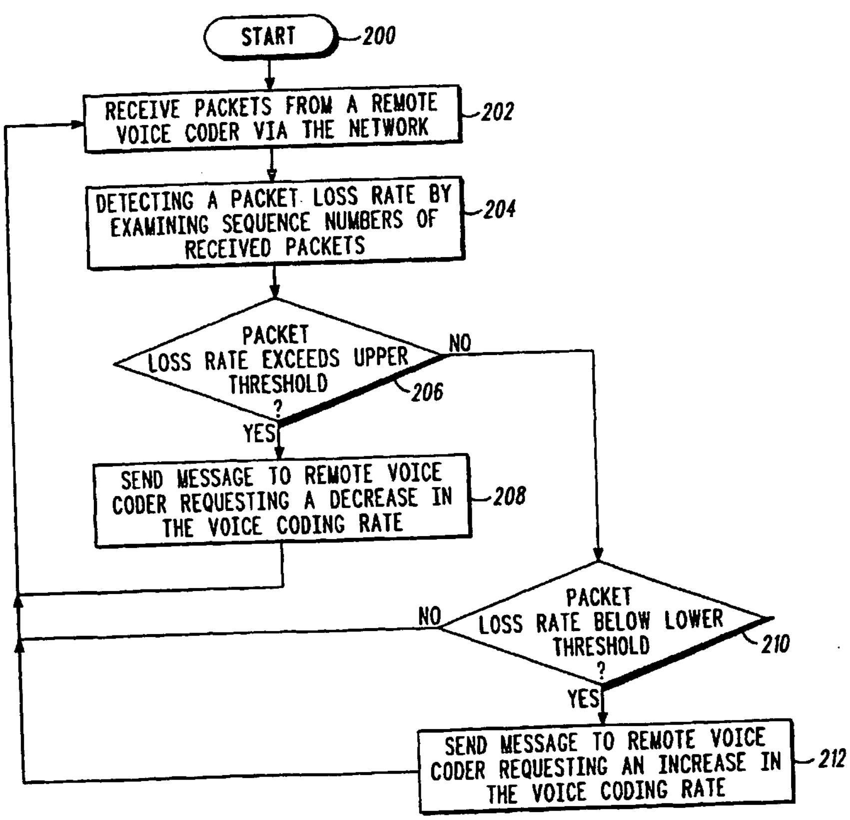
(54) Title: METHOD AND SYSTEM FOR TRANSMITTING AND RECEIVING VOICE PACKETS OVER A COMMUNICATIONS NETWORK



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METHOD AND SYSTEM FOR TRANSMITTING AND RECEIVING VOICE PACKETS OVER A COMMUNICATIONS NETWORK

Background of the Invention

Before voice can be transmitted over a network, the voice must be sampled and encoded to produce data that represents speech. Adaptive multi-rate (AMR) speech codecs represent a new generation of coding algorithms that are designed to work with inaccurate transport channels, such as wireless transmission channels. The AMR speech codec has built-in mechanisms that make it tolerant to a certain level of bit errors introduced by the transport channel. It is designed to restore the original speech, with some degradation, even though the coded speech is received with some bit errors.

In most Internet Protocol (IP) networks, precise data transportation is the norm, and whenever bit errors are detected in the data being transported, the data is discarded. Usually, the transport protocol (e.g., User Datagram Protocol (UDP) or Transmission Control Protocol (TCP)) performs the bit error checking and drops packets that are found with errors.

When transmitting voice data over an IP network the quality of speech at the receiving end may be degraded when network congestion causes voice data packets to be lost or discarded in the network. When the IP network encounters congestion, some routers between a voice packet sender and a voice packet receiver may

receive more data packets than they can timely forward on to their neighboring routers. This will cause the congested router to randomly drop some data packets, which may include the voice packets from the voice packet sender.

Therefore, it should be apparent that there remains a need for an improved methoding system for transmitting and receiving voice packets over a communications network. Once congestion on the network has been detected, the improved method and system should attempt to alleviate the congestion to reduce the number of dropped packets while maintaining voice quality.

Brief Description of the Drawings

The novel features believed characteristic of the invention are set forth in the appended claims. The invention itself, however, as well as a preferred mode of use, further objects, and advantages thereof, will best be understood by reference to the following detailed description of an illustrative embodiment when read in conjunction with the accompanying drawings, wherein:

FIG. 1 illustrates a communications network for transmitting voice packets from a packet sender to a packet receiver in accordance with the method and system of the present invention;

FIG. 2 is a high-level block diagram of a voice packet transceiver in accordance with the method and system of the present invention;

FIG. 3 is a high-level logic flow chart that illustrates the method and operation of receiving a voice packet in accordance with the method and system of the present invention;

FIG. 4 is a high-level logic flow chart that illustrates the method and operation of transmitting a voice packet in accordance with the method and system of the present invention; and

FIG. 5 is a more detailed representation of a voice packet in accordance with the method and system of the present invention.

Detailed Description of the Invention

With reference now to FIG. 1, there is depicted a communications network for transmitting and receiving voice data contained in voice packets. As illustrated, communications network 20 includes packet sender 22 and packet receiver 24 that communicate with one another through IP link 26. IP link 26 is preferably implemented with a network running internet protocol (IP) to route a data packet from its source, such as packet sender 22, to its destination, such as packet receiver 24.

In order to convert speech into data, packet sender 22 includes multi-rate speech encoder 28. Multi-rate speech encoder 28 is preferably implemented with an adaptive multi-rate (AMR) speech coder that is capable of encoding speech bits in a plurality of modes, wherein each mode encodes a different number of speech bits for the same speech input signal. AMR speech coders are more completely described in an article entitled "AMR Speech Codec; General Description (3G TS 26.071 Version 3.0.1)," published by 3rd Generation Partnership Project (3GPP), June 2000.

Encoding rate control 30 in packet sender 22 controls the rate at which multi-rate speech coder 28 encodes speech. Encoding rate control 30 determines an encoding rate based in part upon "change encoding rate messages" sent from packet receiver 24 to packet sender 22 through IP link 26. In a preferred embodiment, these messages request either an increase in speech encoding rate or a decrease in speech encoding rate. Alternately, these messages may

request a specific encoding rate. The change encoding rate message may also be referred to as a "mode request" message.

In order to generate a change encoding rate message, packet receiver 24 includes packet loss monitor 32, which determines whether or not a packet is missing in packet receiver 24, and determines a packet loss rate that indicates a number of packets that have been lost in a selected period of time. Change encoding rate requests are sent from packet receiver 24 in response to the packet loss rate exceeding, or falling below, an upper or lower threshold, respectively.

As speech packets travel through IP link 26, network congestion—a condition similar to rush hour traffic on city streets—may result in some speech packets being dropped. Congestion happens when a network router is overrun with incoming traffic. That is, when data packets arrive before previous packets have been forwarded, the router will have to provide temporary storage to hold them for later forwarding. When too much data in too many data packets arrive before previous data is forwarded, the router may run out of memory, and the router will discard additional packets that arrive during the overflow condition. Packet loss monitor 32 examines serial numbers attached to each packet and determines whether or not a packet is missing at the receiver.

With reference to FIG. 2, there is depicted a high-level block diagram of a voice packet transceiver in accordance with the method and system of the present invention. As illustrated, voice packet transceiver 50 includes adaptive multi-rate (AMR) speech codec 52,

which receives speech input 54 and produces speech output 56. If voice packet transceiver 50 is used, for example, in a cellular telephone, speech input 54 may come from a microphone, and speech output 56 may be sent to a speaker. If voice packet transceiver 50 is used in cellular communications system infrastructure, for example in a base station, speech input 54 and speech output 56 may be coupled to the public switched telephone network (PSTN).

As speech input 54 is encoded, AMR speech codec 52 produces encoded voice bits 58, which are then used to form an encoded voice bit payload 60 portion of adaptive voice rate packet 62.

Encoded voice bits 58 are also coupled to CRC generator 64, which generates a CRC for adaptive rate voice packet 62. The CRC is put into an encoded voice bit header 66 portion of adaptive rate voice packet 62.

In order to control the encoding and decoding rate of AMR speech codec 52, encode rate 68 and decode rate 70 are input into the codec. The values of encode rate 68 and decode rate 70 are determined by rate controller 72.

To produce speech output 56, AMR speech codec 52 receives adaptive rate voice packet 74, which includes encoded voice payload bits 76 and encoded voice bit header 78. As shown, encoded voice bits 84 from encoded voice bit payload 76 are input into AMR speech codec 52. Additional information, such as encoding rate information, comes from encoded voice bit header 78 to control the

decode rate for that fame. Decode rate information 80 is shown coupled to rate controller 72.

Error detector 82 receives speech bits from encoded voice bit payload 76, and a speech bit CRC from encoded voice bit header 78. Error detector 82 then calculates a CRC using encoded voice bits 84, and compares the calculated CRC with the CRC from encoded voice bit header 78. Error detector 82 is coupled to AMR speech codec 52 so that the codec may be informed that a speech bit error has been detected.

According to an important aspect of the present invention, encoded voice bit header 78 includes sequence number 86 that indicates the order in which encoded voice bits 76 were encoded. This order is important because the speech must be decoded in the same frame order used when it was encoded.

Sequence number 86 is coupled to packet loss monitor 32, which is part of the function of rate controller 72. As discussed previously with reference to FIG. 1, packet loss monitor 32 computes a packet loss rate that may represent a number of packets lost over a selected period of time. Alternatively, the packet loss rate may be calculated as a percentage of packets lost out of all the packets sent. It is assumed that the packet loss rate determined by packet loss monitor 32 is relative to the amount of network congestion in the network connected to voice packet transceiver 50.

Rate controller 72 generates information that goes into adaptive rate voice packet 62. Such information includes encode rate 88, which goes into encoded voice bit header 66 to indicate the

rate at which encoded voice bit payload 60 as been encoded. Rate controller 72 also generates a change encoding rate message 90, which is placed in mode request field 92 in encoded voice bit payload 60. Change encoding rate message 90 is used to request that a remote voice coder encode packets at a different encoding rate.

It is important to note that sequence number 86 in an incoming adaptive rate voice packet 74 is used to generate a change encoding rate message 90 in order to alleviate congestion on a heavily congested network, or, alternatively, to take advantage of the bandwidth available on a lightly congested network. If all voice packet transceivers 50 connected to a network request a lower voice encoding rate when heavy network congestion is detected, the network congestion may be alleviated. And, when the network is lightly congested, voice packet transceivers 50 may request higher encoding rates.

According to another aspect of the present invention, change encoding rate message 90 is placed in mode request field 92 in encoded voice bit payload 60, rather than being placed in encoded voice bit header 66. This is important when encoded voice bit header 66 is an error intolerant portion of adaptive rate voice packet 62, and encoded voice bit payload 60 is an error tolerant portion of the voice packet. By placing mode request field 92 in the payload portion of the packet, any errors in change encoding rate message 90 will not be detected by the transport layer of the network, which would cause the transport layer to discard the packet. When using the present invention, errors in the change encoding rate message 90 will not cause the transport layer to discard the packet.

Just as outgoing adaptive rate voice packet 62 includes mode request field 92, incoming adaptive rate voice packet 74 may include mode request field 94 that includes a change encoding rate message 96 that requests a change in encode rate 68. Change encoding rate message 96 may include parity information, which is checked by parity checker 98.

With reference now to FIG. 3, there is a depicted high-level logic flow chart that illustrates the method and operation of receiving a voice packet in accordance with the method and system of the present invention. As illustrated, the process begins at block 200, and thereafter passes to block 202 wherein the process receives packets from a remote voice coder via the network. The remote voice coder is capable of encoding speech bits at various encoding rates. The remote voice coder is also able to receive messages requesting an increase or decrease in voice encoding rates.

Next, the process detects a packet loss rate by examining sequence numbers of received packets, as depicted at block 204. To detect a packet loss rate, the process may examine sequence numbers of received packets and detect that some packets are missing. A packet loss rate represents a number of packets lost in a predetermined period of time, or alternatively, a percentage of packets lost, such as, 10 out of 100 sent.

After detecting a packet loss rate, the process determines whether or not the packet loss rate exceeds an upper threshold, as illustrated in block 206. The upper threshold should be set at a rate that is likely to indicate that the network is heavily congested to a

point that the quality of real-time voice communication is likely to fall below an acceptable level.

If the packet loss rate exceeds an upper threshold, the process sends a change encoding rate message to the remote voice coder to request a decrease in the voice coding rate, as depicted in block 208. The decrease in voice coding rate should decrease the number of bits in each of the future frames sent by the remote voice coder. A decreased number of bits in future frames should contribute to lowering the congestion level of the network. When packets become smaller, the total amount of network traffic will be reduced, which, in turn, will reduce the amount of memory storage required in the congested router, and the router will then become less likely to drop future packets.

After the process sends the decrease coding rate message, the process iteratively returns to block 202 to receive additional packets.

With reference again to block 206, if the packet loss rate does not exceed an upper threshold, the process then determines whether or not the packet loss rate has fallen below a lower threshold, as illustrated at block 210. The lower threshold should be selected to coincide with network congestion falling to a level that would support the transmission of additional voice packets at an acceptable packet loss rate, and hence, a higher voice coding rate would be supported. In some embodiments of the present invention, the upper and lower thresholds may be set to the same value. However, in a preferred embodiment, the upper and lower

thresholds are spaced apart to add some hysteresis, or delay, in the sending of the change encoding rate message.

If the packet loss rate has fallen below the lower threshold, the process sends a change encoding rate message to the remote voice coder to request an increase the voice coding rate, as depicted at block 212. Such an increase in the voice coding rate will increase the voice quality at the receiver, and take advantage of the fact that the network congestion as fallen to a lower level.

If, at block 210, the packet loss rate has not fallen below a lower threshold, the process iterately returns to block 202 to receive the next packet.

As can be seen from the flow chart in FIG. 3, network congestion is detected by calculating a packet loss rate and determining whether or not that rate exceeds a threshold. Messages to change encoding rates at a remote voice coder are sent from the receiver, depending upon the assumed level of network congestion and thresholds set in the receiver.

With reference now to FIG. 4, there is depicted a high-level logic flow chart that illustrates the method and operation of transmitting a voice packet in accordance with the method and system of the present invention. As shown, the process begins at block 300, and thereafter passes to block 302 wherein the process encodes a frame of voice bits. Next, the process generates a CRC for the encoded voice bits, as illustrated at block 304.

After generating the CRC, the process generates a change encoding rate message to request a change of the voice encoding rate in a remote voice coder, as depicted at block 306. The change encoding rate message requests either an increase in coding rate or decrease in coding rate, as determined by a packet loss rate at the remote voice coder, which is described more completely in relation to FIG. 3. The process may also generate parity information that will be added to mode request field 92. At the receiving codec, the parity information can be used to verify the integrity of mode request field 92. If the parity check fails, the codec will ignore the change encoding rate message 90, but the codec will still decode the received speech bits normally.

After generating the change encoding rate message, the process generates an error tolerant portion of an encoded voice packet, wherein the error tolerant portion includes the change encoding rate message and encoded voice bits, as illustrated at block 308. An error tolerant portion of a packet is one that will not be used by the transport layer for determining whether or not to discard the packet. In other words, the transport layer will not perform a checksum-type check on the error tolerant portion to determine whether or not that portion should be passed on through the network.

Referring to FIG. 5, there is depicted a more detailed representation of a voice packet in accordance with the method and system of the present invention. As illustrated, adaptive rate voice packet 62 includes error tolerant portion 100 that includes change encoding rate message 102, and encoded speech bits 104.

After generating an error tolerant portion of the voice packet, the process generates an error intolerant portion of an encoded voice packet, as depicted at block 310 of FIG. 4. The error intolerant portion includes a packet sequence number and a CRC for the encoded voice bits. The error intolerant portion of the encoded voice packet is a portion that is used by the transport layer for determining whether or not to discard the packet. In FIG. 5, error intolerant portion 106 includes real time protocol (RTP) header 108 and AMR frame header 110.

RTP header 108 includes sequence number 86, which tells the receiving codec the order in which to perform frame decoding, and payload type 114, which is used to identify the payload in the RTP as an AMR payload.

AMR frame header 110 includes frame coding rate information 116, which tells the receiving codec the rate to perform speech decoding. Also contained in AMR frame header 110 is speech bit CRC 118, which is used to determine whether or not speech bits 104 contain an error. For more information about RTP protocol, see the article entitled "RTP: A Transport Protocol for Real-Time Applications," RFC1889, published by Internet Engineering Task Force, Jan. 1996.

After generating the error intolerant portion of the packet, the process forms an encoded voice packet by concatenating a voice packet header, the error intolerant portion, and the error tolerant portion, as illustrated at block 312. FIG. 5 shows adaptive rate

voice packet 62, which is formed by UDP header 112, error intolerant portion 106, and error tolerant portion 100.

UDP header 112 includes UDP partial checksum 120, which is used by the transport layer to check error intolerant portion 106 for errors, and if an error is detected, the transport layer will discard the packet. Note that only part of the packet is checked for errors by the transport layer in deciding to discard a packet—error tolerant portion 100 is not checked for errors by the transport layer. For more information regarding UDP, see the article entitled "User Datagram Protocol", which is RFC768 published by Internet Engineering Task Force (IETF) August 1980.

In current wireless cellular communications system design, referred to as a 3GPP system, an 8-bit CRC field is used for detecting errors in a more sensitive portion of the speech bits in a voice packet—a portion referred to as Class A bits in an AMR frame. But this CRC field is only added by the radio transmitter before the AMR frame is sent over the air link, and it is removed by the radio receiver right after the frame is received from the air link. In other words, the CRC mechanism, as defined in 3GPP, is only applied to the over-the-air link, and is not available for use in any of the other links of the voice over IP (VoIP) connection.

According to another aspect of the present invention, an 8-bit CRC field is added to the AMR frame format as a permanent field—a field that will remain in the frame all the way to the AMR decoder, where it will be examined by the decoder. The AMR encoder will generate the CRC, rather than being generated by the

radio transmitter in the middle of the connection, and the CRC will then be used by the receiving AMR decoder, rather than being removed by the radio receiver in the middle of the connection. The advantages of this method and system include: 1) simplifying the wireless transport layer at the transmitter and receiver in the radio link because the wireless transport layer will no longer need to understand the format of the payload frame (in the prior art 3GPP approach the transport layer needs to know where to find the Class A bits, etc); 2) error checking the sensitive Class A bits all the way through the entire connection.

The foregoing description of a preferred embodiment of the invention has been presented for the purpose of illustration and description. It is not intended to be exhaustive or to limit the invention to the precise form disclosed. Obvious modifications or variations are possible in light of the above teachings. The embodiment was chosen and described to provide the best illustration of the principles of the invention and its practical application, and to enable one of ordinary skill in the art to utilize the invention in various embodiments and with various modifications as are suited to the particular use contemplated. All such modifications and variations are within the scope of the invention as determined by the appended claims when interpreted in accordance with the breadth to which they are fairly, legally, and equitably entitled.

Claims

What is claimed is:

1. A method in a network for improving transmitted voice quality, the method comprising the steps of:

receiving packets from a voice coder, wherein the packets have been encoded at a higher encoding rate;

detecting a packet loss rate that exceeds a threshold loss rate;

sending a reduce encoding rate message to the voice coder to cause the voice coder to encode the packets at a lower encoding rate.

2. The method for improving transmitted voice quality according to claim **1** wherein the step of sending a reduce encoding rate message to the voice coder further includes sending a reduce encoding rate message to the voice coder in an error tolerant portion of a packet, wherein the error tolerant portion is not used by the transport layer for determining whether or not to discard the packet.

3. The method for improving transmitted voice quality according to claim **2** wherein the step of sending a reduce encoding rate message to the voice coder in an error tolerant portion of a packet further includes sending a reduce encoding rate message and message parity information to the voice coder in an error tolerant portion of a packet, wherein the message parity information is generated from the reduce encoding rate message.

4. The method for improving transmitted voice quality according to claim **1** further includes verifying the integrity of selected voice bits in a received packet by calculating a cyclic redundancy check value for the selected voice bits and comparing the cyclic redundancy check value to an included cyclic redundancy check value that was put in the received packet at the voice coder.

5. The method for improving transmitted voice quality according to claim **4** wherein the selected voice bits are class A voice bits in an adaptive multi-rate encoded frame.

6. The method for improving transmitted voice quality according to claim **1** wherein the step of receiving packets from a voice coder further includes receiving packets of adaptive multi-rate encoded voice bits from a voice coder.

7. A system in a network for improving transmitted voice quality comprising:

means for receiving packets from a voice coder, wherein the packets have been encoded at a higher encoding rate;

means for detecting a packet loss rate that exceeds a threshold loss rate;

means for sending a reduce encoding rate message to the voice coder to cause the voice coder to encode the packets at a lower encoding rate.

8. The system for improving transmitted voice quality according to claim **7** wherein the means for sending a reduce encoding rate message to the voice coder further includes means for sending a reduce encoding rate message to the voice coder in an error tolerant portion of a packet, wherein the error tolerant portion is not used by the transport layer for determining whether or not to discard the packet.

9. The system for improving transmitted voice quality according to claim **8** wherein the means for sending a reduce encoding rate message to the voice coder in an error tolerant portion of a packet further includes means for sending a reduce encoding rate message and message parity information to the voice coder in an error tolerant portion of a packet, wherein the message parity information is generated from the reduce encoding rate message.

10. The system for improving transmitted voice quality according to claim **7** further includes means for verifying the integrity of selected voice bits in a received packet by calculating a cyclic redundancy check value for the selected voice bits and comparing the cyclic redundancy check value to an included cyclic redundancy check value that was put in the received packet at the voice coder.

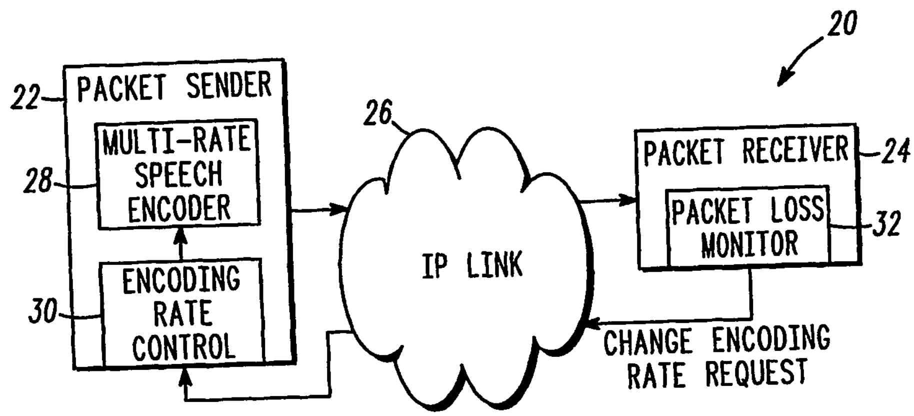
20
22
PACKET SENDER
28
MULTI-RATE SPEECH ENCODER
30
ENCODING RATE CONTROL
26
IP LINK
PACKET RECEIVER
24
PACKET LOSS MONITOR
32
CHANGE ENCODING RATE REQUEST

FIG. 1

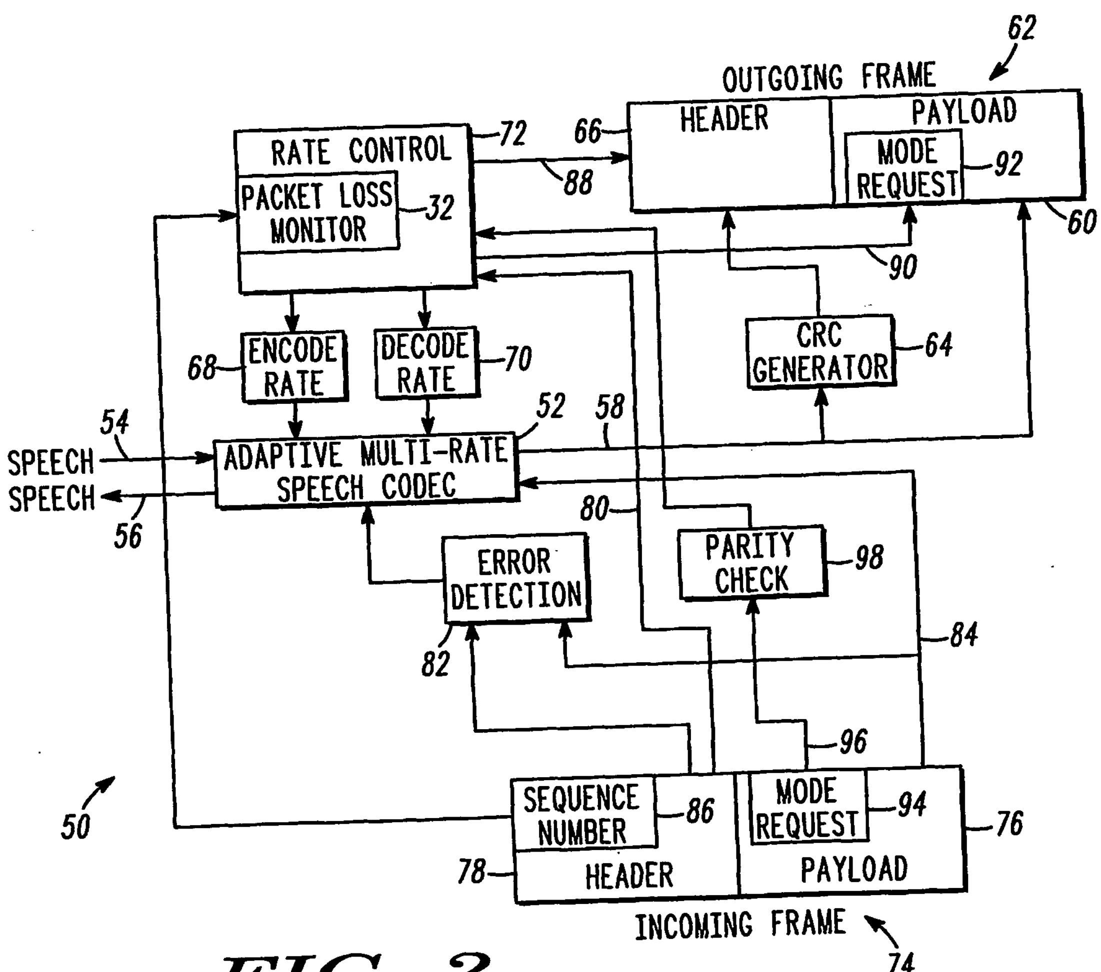
62
OUTGOING FRAME
HEADER
PAYLOAD
66
MODE REQUEST
92
60
RATE CONTROL
72
88
PACKET LOSS MONITOR
32
90
CRC GENERATOR
64
68
ENCODE RATE
DECODE RATE
70
54
52
58
SPEECH
ADAPTIVE MULTI-RATE SPEECH CODEC
SPEECH
56
80
PARITY CHECK
98
ERROR DETECTION
82
84
96
50
SEQUENCE NUMBER
86
MODE REQUEST
94
76
78
HEADER
PAYLOAD
INCOMING FRAME
74

FIG. 2

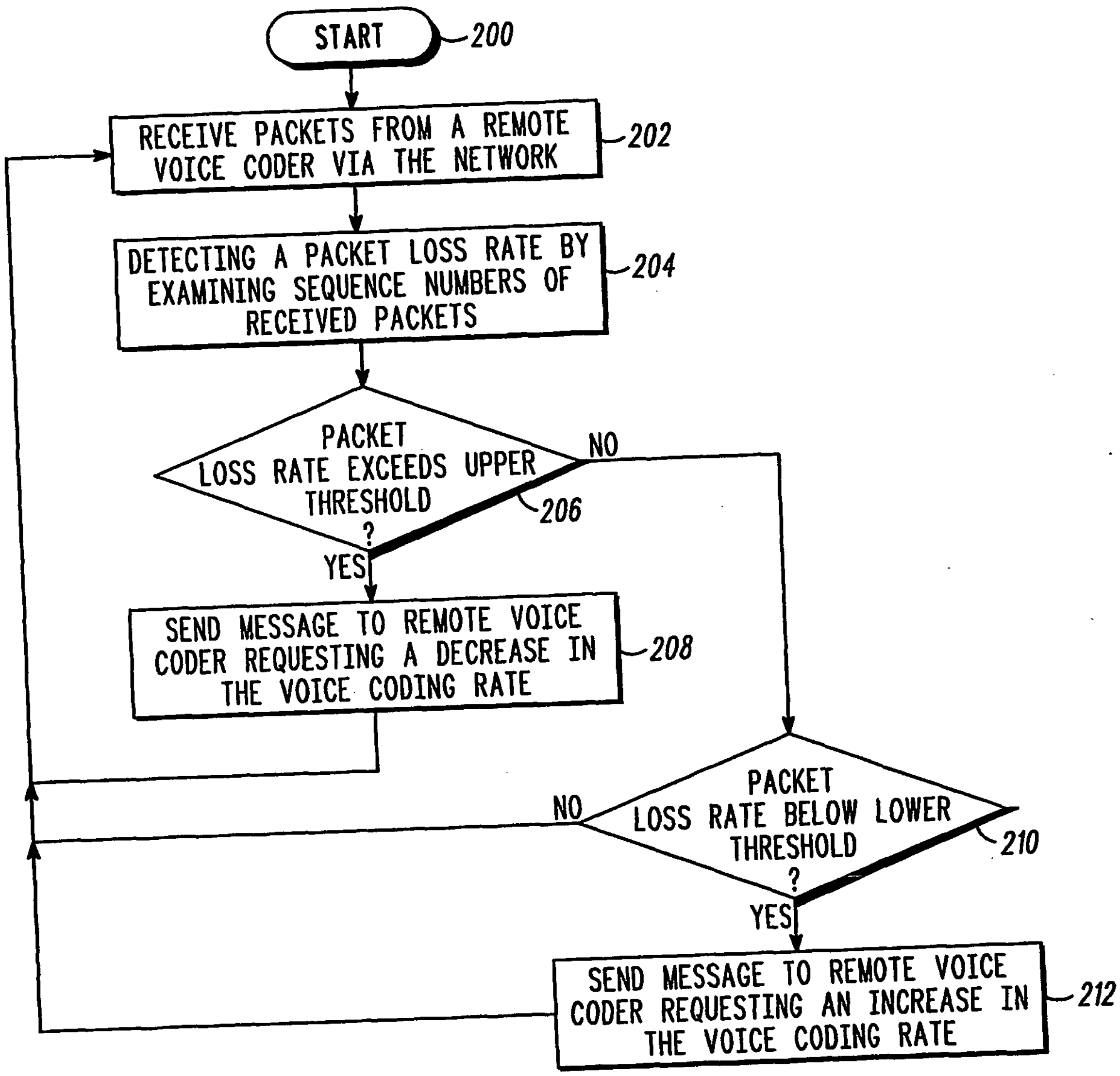
START
200
RECEIVE PACKETS FROM A REMOTE VOICE CODER VIA THE NETWORK
202
DETECTING A PACKET LOSS RATE BY EXAMINING SEQUENCE NUMBERS OF RECEIVED PACKETS
204
PACKET LOSS RATE EXCEEDS UPPER THRESHOLD ?
206
NO
YES
SEND MESSAGE TO REMOTE VOICE CODER REQUESTING A DECREASE IN THE VOICE CODING RATE
208
PACKET LOSS RATE BELOW LOWER THRESHOLD ?
210
NO
YES
SEND MESSAGE TO REMOTE VOICE CODER REQUESTING AN INCREASE IN THE VOICE CODING RATE
212

FIG. 3

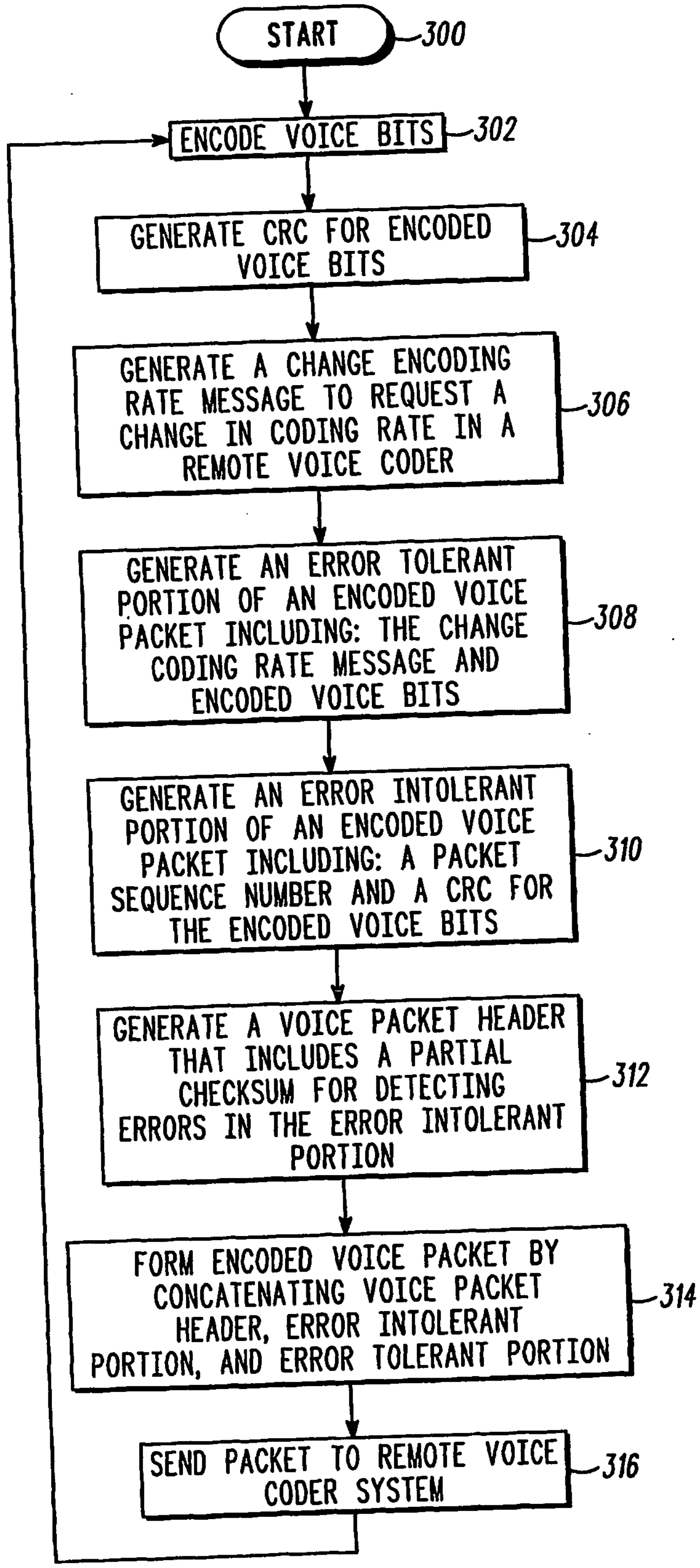
START
300
ENCODE VOICE BITS
302
GENERATE CRC FOR ENCODED VOICE BITS
304
GENERATE A CHANGE ENCODING RATE MESSAGE TO REQUEST A CHANGE IN CODING RATE IN A REMOTE VOICE CODER
306
GENERATE AN ERROR TOLERANT PORTION OF AN ENCODED VOICE PACKET INCLUDING: THE CHANGE CODING RATE MESSAGE AND ENCODED VOICE BITS
308
GENERATE AN ERROR INTOLERANT PORTION OF AN ENCODED VOICE PACKET INCLUDING: A PACKET SEQUENCE NUMBER AND A CRC FOR THE ENCODED VOICE BITS
310
GENERATE A VOICE PACKET HEADER THAT INCLUDES A PARTIAL CHECKSUM FOR DETECTING ERRORS IN THE ERROR INTOLERANT PORTION
312
FORM ENCODED VOICE PACKET BY CONCATENATING VOICE PACKET HEADER, ERROR INTOLERANT PORTION, AND ERROR TOLERANT PORTION
314
SEND PACKET TO REMOTE VOICE CODER SYSTEM
316

FIG. 4

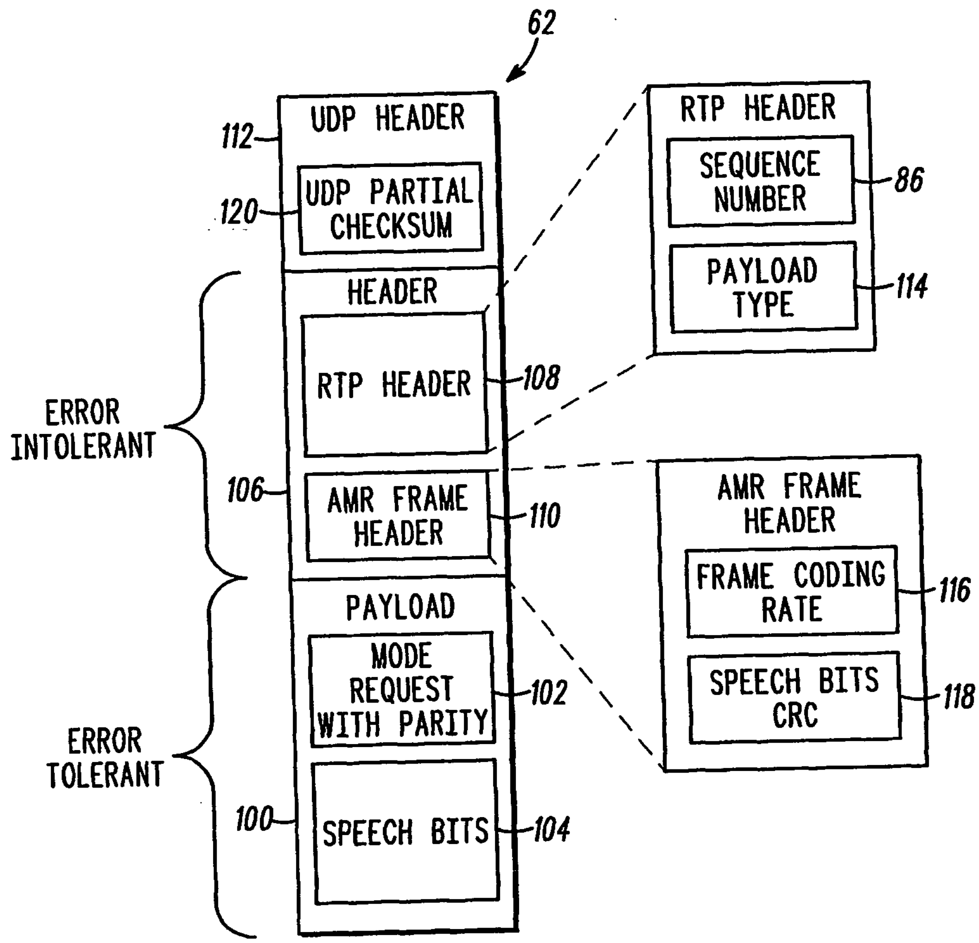
62
UDP HEADER
112
UDP PARTIAL CHECKSUM
120
HEADER
RTP HEADER
108
AMR FRAME HEADER
110
106
ERROR INTOLERANT
PAYLOAD
MODE REQUEST WITH PARITY
102
SPEECH BITS
104
100
ERROR TOLERANT
RTP HEADER
SEQUENCE NUMBER
86
PAYLOAD TYPE
114
AMR FRAME HEADER
FRAME CODING RATE
116
SPEECH BITS CRC
118

FIG. 5